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Headphone Acoustic Measurement and Quality Control

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The study documented in this thesis was commissioned by Hefio Oy. It was conducted as a part of the product development of their Hefio Play headphones. The main objective of the study was the proof of concept of a prototype acoustic measurement system. The measurement system is designed to be used for quality control and factory calibration of the mass-produced headphones. The study was focused on the quality control aspect of the measurement system.

The first half of the thesis discusses headphones, headphone measurements and statistical quality control in general. The scope of those three broad topics has been limited to the information necessary for this study.

The second half of the thesis discusses the equipment used, measurements conducted, and the analysis of the measurement data. All of the measurements for the study were conducted on eight prototype headphones.

The measurement data was analysed using statistical methods. The analysis ends up with three indicators of quality: average sensitivity, average response and measurement accuracy.

The study succeeded in the proof of concept of the measurement method. Larger and more consistent sample is needed to evaluate the precision of the measurements accurately.

Keywords	headphones, acoustic measurement, quality control, statistics

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Tämä työ toteutettiin tilauksena Hefio Oy:lle ja se oli osa Hefio Play kuulokkeen tuotekehitystä. Työn tavoitteena oli testata demovaiheessa olevaa kuulokkeiden akustista mittajärjestelmää. Tähän perustuvaa mittajärjestelmää on tarkoitus käyttää Hefion massatuotettujen kuulokkeiden laadunvalvontaan, sekä tehdaskalibrointiin. Tähän opinnäytetyöhön raportoitu osuus on keskittynyt laitteen laadunvalvonta käyttöön.

Opinnäytetyön alkupuolisko käsittelee kuulokkeita, kuuloke mittauksia ja tilastollista laadunvalvontaa yleisesti. Näistä kolmesta laajasta aiheesta on valikoitu käsiteltäväksi tietoa, joka tukee työn loppupuoliskoa.

Toinen puolisko pitää sisällään laitteiston, mittaustavan ja mittaustulosten analysoinnin. Työtä varten suoritettiin impulssivastemittauksia kahdeksalle prototyyppikuulokkeelle.

Mittaustulokset analysoitiin tilastollisia menetelmiä käyttäen. Analyysi päätyi kolmeen laadun indikaattoriin, jotka ovat: keskimääräinen herkkyys, keskimääräinen vaste ja mittaustarkkuus.

Työ onnistui todentamaan mittaustavan toimivuuden. Mittaustarkkuuden arvioimiseksi tarvitaan suurempaa ja tasalaatuisempaa näytettä.

Avainsanat	kuulokkeet, akustiset mittaukset, laadunvalvonta, tilastotiede

Contents

Abbreviations

ADC	Analog-to-digital converter
ANC	Active noise cancellation
BA	Balanced armature
CTQ	Critical-to-quality
DAC	Digital-to-analog converter
DSP	Digital signal processor or digital signal processing
FIR	Finite impulse response
hi-fi	High fidelity
HPF	High-pass filter
IEM	In-ear monitor
IIR	Infinite impulse response
LCL	Lower control limit
LPF	Low-pass filter
SNR	Signal-to-noise ratio
SPC	Statistical process control
SPL	Sound pressure level
SQC	Statistical quality control
THD	Total harmonic distortion
UCL	Upper control limit

1 Introduction

This bachelor's thesis is a part of the product development of Hefio Play headphones for Hefio Oy. Hefio Oy is a company focusing on technology for individual calibration of headphones (see section 2.6.2). The goal of this study is to present the proof of concept of an acoustic measurement system. The measurement system will eventually be used for pass/fail testing and for collecting statistical data of the acoustic quality characteristics of the mass-produced Hefio Play headphones.

The measurements for this study were conducted on the early hand-built protypes earphones assembled from 3D-printed parts. The Measurement system used in the measurements is also the first prototype built for the proof of concept of the measurement method. The purpose of the study is to lay down the ground work for the further development of the measurement system to be used in the factory environment as an integral part of the mass-production process.

Chapters 2 to 4 of this thesis discuss theory behind headphones, acoustic measurements and statistical quality control to back up the chapters 5 to 7. The chapters 5 to 7 present the actual measurement system, as well as the measurement data and its analysis conducted for the purposes of this study. Despite this thesis being written with a specific product in mind, it also gathers a fairly broad and general set of information about headphones, acoustical measurements and quality control.

2 Headphones

In this chapter the concept of headphones is introduced. Most common applications and various types of headphones and their properties are discussed and visualized in figures. The aim of this chapter is to provide necessary theory to back up the later chapters of this thesis, as well as to categorize the Hefio Play headphones based on the presented types and properties. The emphasis of this chapter is on the features that are present in the Hefio Play headphones.

2.1 Uses for Headphones

Music has become portable and easier to access, starting from the first Walkman cassette player in 1979, mp3 players, and finally music streaming services like Spotify combined with smartphones. The latest has granted music listeners almost endless pool of music anytime anywhere. Insert type headphones and earbuds are the most portable type and most convenient to use on the go with a smartphone. Wireless headphones take portability and convenience even further by connecting wirelessly to a smartphone.

Headphones for music professionals and audiophiles need to constantly push the boundaries of audio reproduction. A music consumer might enjoy either emphasized low or high frequencies and be willing to trade of sound quality over portability. But for reference purposes, e.g., mixing, the headphone should be as “transparent” as possible. This means the headphone should reproduce spectral content as it is without any coloration. Most high-end headphones are over the ear type with either closed or open back design, making them less portable. There are some Hi-Fi in-ear monitor (IEM) options available. Special category in the IEM type is the custom in-ear monitor (CIEM) which has a customised ear piece to fit perfectly the listener’s ear shape.

Another area where headphones are widely used is gaming. Especially first-person point of view games can become more immersive when combined with well-designed headphones. This is where 3D-headphones with head tracking capability would benefit a gamer a lot, making the gaming experience more immersive. Gaming headphones tend to be the over ear and closed back type combined with a microphone for headset functionality.

2.2 Definition

According to Audio Precision [1] headphone is defined as following.

> IEC 60268-7 [1] defines earphone as an electroacoustic transducer intended to be closely coupled to the ear, headphone as an assembly of one or two earphones which may or may not be mounted on a headband and headset as a headphone assembly equipped with a microphone.

The definition shows two major aspects; earphone being an electroacoustic transducer, and the coupling of the earphone to the ear. The coupling comprises not only the acoustic coupling of the transducer to the ear canal but also the acoustic coupling of the external world to the ear canal. Types and properties of these two aspects are introduced in Chapters 2.3 and 2.4.

2.3 Coupling to the Ear

The most common coupling types between earphone and ear canal are illustrated in Figure 1 below. On the upper row are also demonstrated three different acoustical coupling types between the ear canal and the external world. The closed-type aims to isolate the ear canal from external noise and the open type provides an intentional leak from the external space into the ear canal. The open type is considered more natural sounding due to environmental acoustic "awareness" [2, 452].

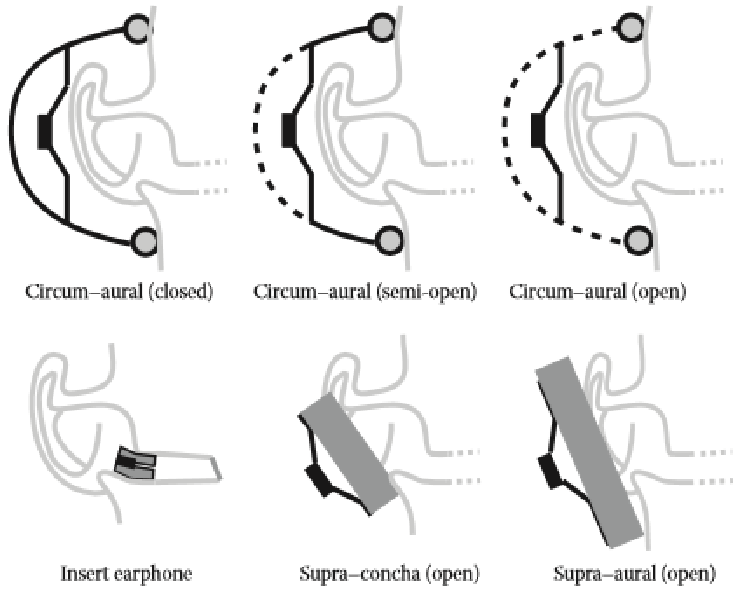

Figure 1 - Headphone Coupling Types. Reprinted from Kleiner [2,452].

2.3.1 Insert Earphone

Insert earphone or in-ear monitor (IEM), illustrated on the lower left corner of Figure 1, is plugged inside the listener’s ear canal. They are usually equipped with rubber or foam tips which is the part that forms a seal between the earphone and the ear canal. When the tip of the earphone fits well and is placed well inside the users ear large amount of passive noise attenuation is achieved. This attenuation is however frequency dependent and better on the higher frequencies. [3.]

The design of insert type headphone must compensate for the changes in the acoustics of the outer ear, that occur when the ear canal is blocked by the earphone [3,3]. Insert IEM headphones are known to have great performance especially on at the high frequencies. They are coupled close to the eardrum which makes the air volume between the transducer and the eardrum small. This combined with the good seal makes producing lower frequencies possible even with a small driver. [3.]

2.3.2 Circumaural

Circumaural aka Around-the-Ear earphones illustrated on the upper row on Figure 1 are the biggest in size. They surround the whole pinna and form a seal against the side of the head around the pinna. Because of this relatively good seal that leaks only due to hair between the head and the cushion compared with relatively large drivers, circumaural headphones are known to have a good bass response. [4, 500] Most studio monitor headphones and reference headphones are circumaural.

2.3.3 Supra-aural and Supra-concha

Supra-aural earphones are designed to rest on the pinna and are the second biggest in size. A good bass response can be produced but it takes more effort than with circumaural headphones. Supra-concha earphones, as seen from the Figure 1, are very similar with the supra-aural ones but they are smaller in size. [4, 500.]

2.3.4 Intra-concha

Intra-concha (not shown in Figure 1) is a type of headphone that are common to get as a tie in with a mobile- or smartphone. Intra-concha type headphones are resting against the ear canal opening but not inserting it. This type of headphone has poor reproduction of the frequency spectrum especially on the lower frequencies due to poor coupling [3].

2.4 Transducer Types

Transducer is the component in a headphone that transfers electric signals (e.g. music) to sound waves. They are commonly called headphone drivers. Headphone drivers come in a few different types with different structures and operation to produce these sound waves. Hefio Play headphones are designed with dynamic transducers so the focus of this section will be on the dynamic type transducers.

2.4.1 Dynamic Driver

The basic operation of a dynamic type driver is explained in this chapter. Dynamic driver aka moving coil driver is the most common transducer type in headphones. It is a miniature version of a typical speaker driver. Dynamic driver translates electronic signals to equivalent acoustic signals using an electric motor like structure combined with a thin diaphragm [5].

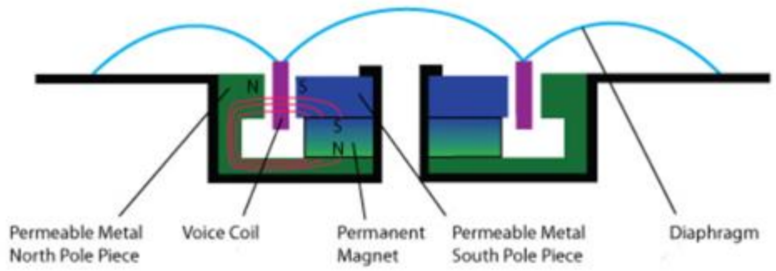

Figure 2 - Cross cut illustration of a dynamic headphone driver. Reprinted from [5].

Figure 2 above illustrates the components of a dynamic driver. The diaphragm is the part responsible of the air movement resulting in audible sound waves. In operation, the diaphragm is moving in and out with the frequency of the electric signal. In Figure 2 the lobe in the middle of the diaphragm is called dome and the side lobes are called flexures. [5.]

As seen from Figure 2, the voice coil is connected right under the diaphragm and around its dome. When current is fed into the voice coil, a magnetic field is created around the wire of the coil. Since the magnetic flux of the permanent magnet is directed from the north pole towards the south pole, according to Fleming's left-hand rule for motors, the voice coil in Figure 2 moves either up or down depending on the direction of the current. The up and down movement of the voice coil is moving in turn the diaphragm attached to it. [5.]

When the diaphragm is moving, soundwaves start to propagate not only outwards from the driver, but also behind the driver getting trapped inside the structure of the driver. This creates acoustic resonances that interfere with the normal movement of the diaphragm. The resonances behind the driver produce acoustic impedances at the resonant frequencies. Quick changes in the acoustic impedance result in peaks in harmonic distortion at the corresponding frequencies. To reduce the distortion, the acoustic resonances need to be controlled. Both acoustic vents, being holes or acoustically transparent materials, and damping are used to reduce the amount of unwanted resonance. [5.] Loudspeaker non-linearities are discussed in more detail by Klippel in his convention paper [6].

2.4.2 Other Types of Transducers

In addition to the common dynamic drivers there are other viable options to use in headphones. This section introduces briefly other types of transducers common in headphones.

a) Balanced Armature (BA)

The balanced armature (BA) drivers are the smallest in size of all the transducer types. This makes it possible for the diaphragm to be made light weight, resulting in better transient response. [7.]

b) Planar Magnetic

The planar magnetic transducer aka isodynamic transducer is acoustically less complicated than the dynamic transducer. It is structured so that the coil is etched on the diaphragm itself which is located between two arrays of magnets. The structure of the planar magnetic driver ensures the driving force is constant on the membrane area. This is in turn results in good linearity and thus in low distortion. [4,502.]

c) Electrostatic and Electret

Electrostatic transducer utilizes a charged membrane. The membrane's charge has to be maintained using a high-voltage DC-supply. The movement of the membrane is achieved with an electric field in which the charged membrane is placed. The audio signal is used to modulate the electric field. [4,508.] Electret transducer is very similar to the electrostatic transducer. The only difference is that the membrane of electret transducer is permanently polarized. [4,510].

2.5 Signal Processing Features

Computers have gotten and are still getting less expensive and more powerful and they come in smaller and smaller packages. At the same time specialized microprocessors for the computational needs of digital signal processing (DSP) have become available. Due to the development of the DSP chips, it has become feasible to implement all sorts of digital functions also to headphones. In this chapter two relatively common signal processing features are introduced. Those features are called the Active Noise Cancellation (ANC) and Digital Crossover.

2.5.1 Active Noise Cancellation

The most notable signal processing feature in the market is the active noise cancellation (ANC). The purpose of ANC is to attenuate ambient noise. ANC chips come with three topologies: feedback feedforward and hybrid. The difference between the three topologies is in the microphone placement. In feedforward, the microphone is placed outside the headphone, giving it more time to process the incoming noise signal. In feedback, the microphone is placed inside the headphone in front of the headphone driver. Finally, the hybrid type is a combination of the feedback and feedforward types with microphones both inside and outside the headphone's structure. [8.]

Active noise cancellation in a headphone can be implemented in simplified terms with a microphone, speaker, battery and the noise cancellation integrated circuit. The principle of operation for a feedforward type ANC is relatively simple. When activated, the microphone or microphones located outside of the headphone are recording the ambient noise of the listening environment close to the user's ears. The recorded noise is then input to the noise cancellation IC which then creates 180º phase shifted version of the recorded noise. Finally, when the phase shifted noise recording is output to the headphone’s drivers, it cancels out the ambient noise in real time. Technical term for the cancellation is destructive interference. [8.]

The current active noise cancellation systems are capable of reducing only low frequency noise. So, the actual structure of the headphone must still offer the passive noise reduction of the higher frequencies. [8.] As an example of the performance of a feedforward active noise cancellation system, the IC AS3415 form AMS will be used as reference. According to the datasheet [9] of the AS3415 the bandwidth for an effective cancellation is from 20 Hz up to 2 – 3 kHz and the amount of achievable reduction is promised to be 25 dB or more.

2.5.2 Crossover

Loudspeaker systems are designed to produce sound with smooth frequency response and low distortion over the intended radiation area. It is not possible to achieve all that with a single driver. That is why multi driver systems are used. Each of the drivers are purposed to reproduce a certain frequency band. A common structure is a two-way system which has a driver called woofer for lower frequencies and a driver called tweeter for the higher frequencies. A three-way system in turn has separate high, middle and low frequency drivers.

To split the full audio spectrum to the separate bands, an electrical crossover network is used. For a two-way-system This crossover network can be implemented in analog domain using both passive and active components, as well as in digital domain using DSP. There are two main types of crossover networks; high-level and low-level. [4,197-198] These two types are illustrated in Figure 3 and Figure 4.

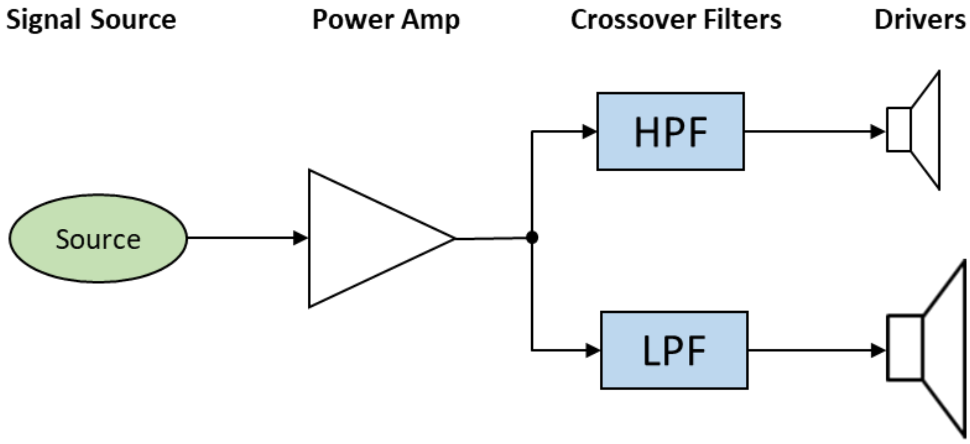

Figure 3 - Crossover network implemented with high-level filters for a two-way system.

a) High-level crossover

High-level crossover network is implemented with passive filters that are driven with the same power amplifier. The filtering is thus done for a high-power signal. Figure 3 shows the amplified signal fed to the tweeter going through a high-pass filter (HPF) and the amplified signal for the woofer going through a low-pass filter (LPF). The output from the source in Figure 3, Figure 4, and Figure 5 is an analog signal. [4,197-198.]

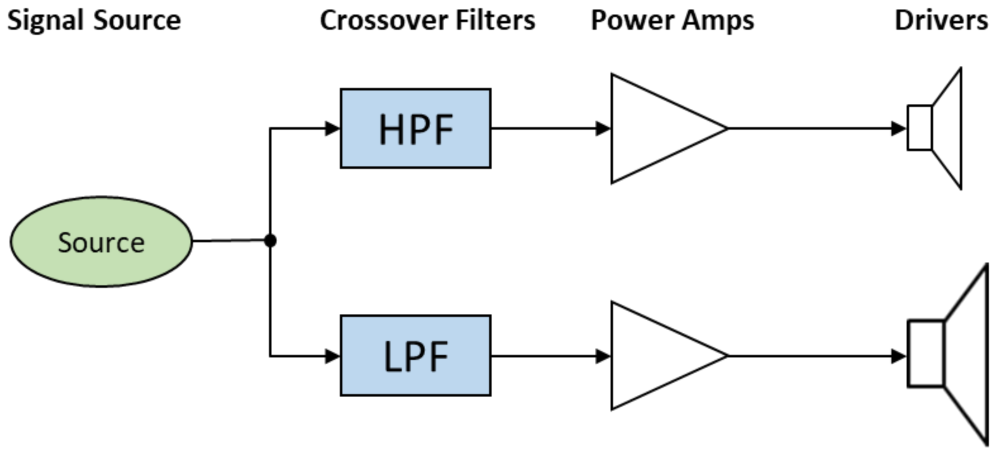

Figure 4 - Crossover network implemented with low-level filters for a two-way system.

b) Low-level crossover

Low level crossover network, illustrated in Figure 4, is implemented using active filters. The filtering is thus done for a low-power or line-level signal. The outputs of the filters are connected to separate power amplifiers for each of the drivers. Figure 4 shows the tweeter being fed with a high-passed signal through its own power amp and the woofer being fed with a low-passed signal through its own power amplifier. [4,198].

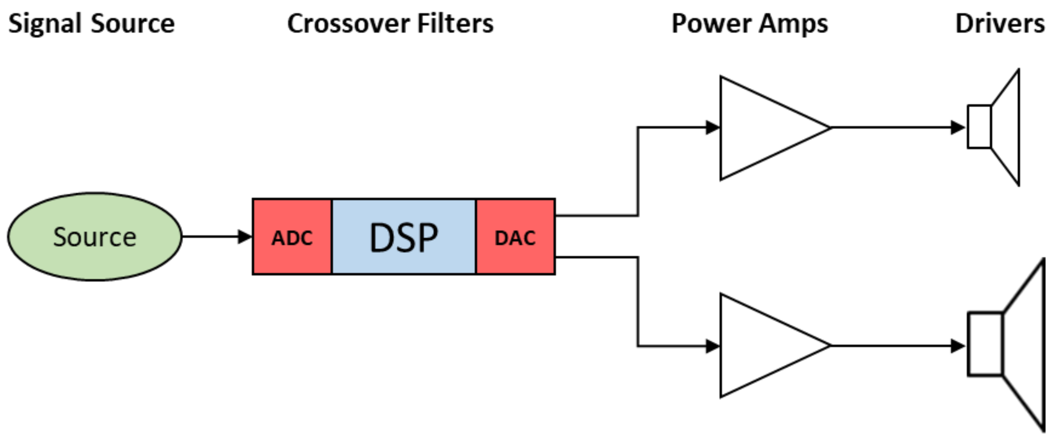

Figure 5 - Digital low-level crossover network for a two-way system.

c) Digital crossover

Instead of the physical filter components, it is possible to implement the crossover in digital domain for a low-level signal. This makes the signal path more complex as the signal needs to be converted between analog and digital domains. In Figure 5 the analog signal from the source is converted to digital domain using an analog-to-digital-converter (ADC). The crossover filters are implemented in the DSP for the digital signal. The signal is then converted back to analog domain with a digital-to-analog-converter (DAC) and send to the power amps of each driver. In some cases, the output of the source can be a digital signal in which case the analog-to-digital-converter is not needed.

The filters of a digital crossover system are implemented in software. The most common digital filter types are the infinite impulse response (IIR) and the finite impulse response (FIR) filters. The software filters can either simulate classical hardware filters or be completely synthesized. With digital crossover, the filters can achieve more complexity without enlarging the filters' physical size. The only limiting factor is the computational power of the DSP-unit. [10.]

Digital crossover implemented on two-way-headphone would be highly beneficial due to its customizable nature. Each produced headphone unit could have individually calculated crossover filter parameters to compensate the manufacturing tolerances of the drivers. The data for these parameters is obtainable form the headphone's frequency response measurements.

2.6 Hefio Play Headphones

The Hefio Play headphones, central to this thesis, are presented in more detail in this chapter.

2.6.1 Description

Hefio Play headphones are functioning in the digital domain and are integrated with a DSP unit which takes care of the signal processing features (Digital Crossover) described in Chapter 2.4. The DSP unit is also responsible for the user specific calibration, to be introduced in chapter 2.6.2. Figure 6 shows an illustrative drawing of the Hefio Play headphones. The DSP unit is not visible in Figure 6.

Figure 6 - Illustrative drawing of the Hefio Play headphones by Hefio Oy.

These headphones are basically insert type earphones. Although they are designed to rest partially on the pinna like the supra-concha earphones the actual path for the sound is inserted in the ear canal. When it comes to transducers they have two separate dynamic drivers; a tweeter for higher frequencies and a woofer for lower frequencies.

In addition to the dual drivers the headphones are also equipped with microphones. One set of microphones is used to implement the basic headset-functions. These microphones are located inside each earphone and angled to pick up signals outside the chassis of the earphone. The second set of microphones is used for the individual calibration of the headphones. These microphones are located near the sound hole of each earphone and are designed to pick up signals inside the ear canal.

2.6.2 Individual Calibration

The digital crossover in an in-ear-type headphone is already something that has never been done before. But the truly unique feature of the Hefio Play headphone is the self-calibration function. The headphone has microphones situated close to the sound hole which is inserted into the user's ear. During the calibration process a measurement signal is played back and the inner microphone records the response inside the user's ear. That response is then processed in the DSP unit to individually produce a pre-defined target spectrum at the eardrum. This includes cancelling out the effect of the user's ear canal. Doing this assures that the sound spectrum at the eardrum is exactly same for each user, even though every user's ear canal is basically acting as a unique filter that distorts the audio away from what it was intended to sound.

3 Headphone Measurements

Ideal headphones would playback audio as it is without any coloration. However, in practise this is not possible, and the headphones will always add coloration to the audio signal. That is why headphone measurements are needed to determine the quality of the reproduction of the spectral content. Headphone measurements, like all electro-acoustic transducer measurements, are trying to find out two things: the linear transfer function of the system via Impulse response measurement and the non-linear behaviour by measuring the non-linear distortions of the system. The simultaneous measurement of linear impulse response and harmonic distortion by using the swept-sine technique is presented by Angelo Farina [11] and the approach is used as the basis for the measurement system presented in this thesis.

3.1 Frequency Response

It is important to address that the frequency response measurement presents only the linear response of a system. In practise, headphone, nor any electro-acoustic transducer is a completely linear system. This approximation is only made to analyse the linear and non-linear components of the system separately.

Frequency response is measure of the output of a system as a function of frequency that describes the dynamics of the system. It is obtained from the impulse response (time domain) by taking a Fast Fourier Transform (FFT) which converts the response from time domain to frequency domain. The response is constructed of two parts; the magnitude and phase compared to the input. For headphones, the input of the system is the input signal fed into the headphone and the output of the system is the acoustic signal picked up by the measurement microphone. In case of acoustic transducers, the output and input are usually presented in sound pressure level (SPL). Frequency response is considered the most influential factor to the perceived sound quality.

Farina’s swept sine technique [11,6] uses a logarithmic sine sweep as the input signal. Logarithmic sine sweep, or log sweep, is a sine wave which frequency is exponentially varied over time. The equation for Farina’s log sweep [11,6] is shown in Equation 1. For headphone measurements, rational range for the swept frequencies is from 20 Hz – 20 000 Hz (f_1 – f_2 in Equation 1) as this is the maximum range for human hearing.

$$x(t) = sin\left[\frac{f_1 \times T}{ln\left(\frac{f_2}{f_1}\right)} \times \left(e^{\frac{t}{T} \times ln\left(\frac{f_2}{f_1}\right)} - 1\right)\right] \tag{1}$$

f_1 is the start frequency

f_2 is the stop frequency

T is the total sweep duration.

Using a logarithmic sine sweep as the measurement signal allows one to separate the harmonic distortion responses from the linear response. A longer sweep results in a good signal-to-noise ratio (SNR). [11,6.]

3.1.1 Headphone Target Responses

The “shape” of the frequency response curve is used to describe the components of the electroacoustic transfer chain. Microphones, amplifiers, playback and recording devices, loudspeakers and headphones are examples of electroacoustic components. Apart from headphones, all the other components in the transfer chain are usually designed to have a flat response, as seen in Figure 7.

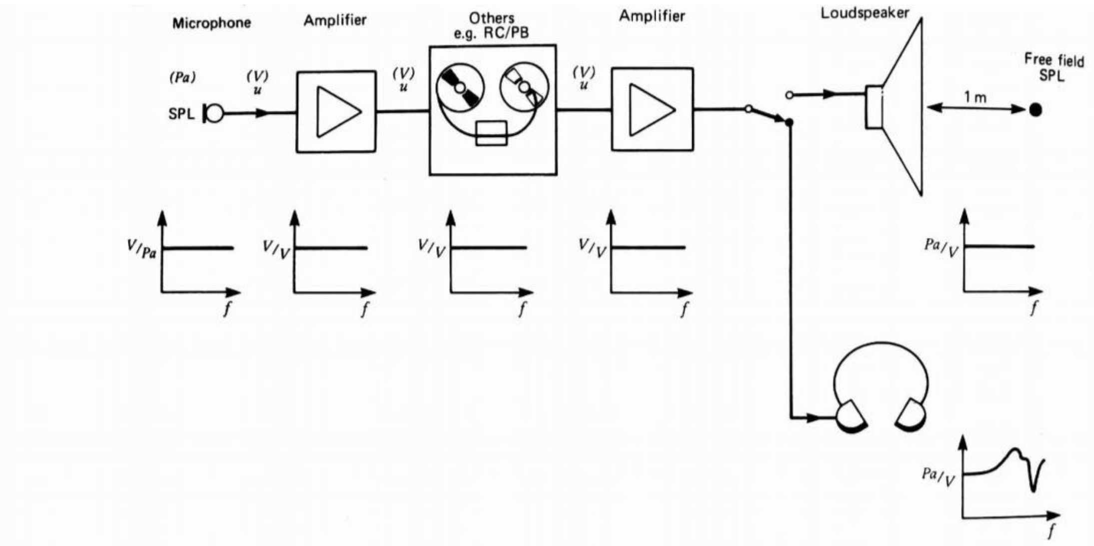

Figure 7 - Block diagram comparing headphones with other components in the Hi-Fi chain. Reprinted from Borwick [4, 494].

It is natural to compare headphones to speakers. Speakers are designed to reproduce recorded sound in a room of sorts. This room affects the perceived sound alongside the listeners head and torso and direction of the sound coming from the speakers. Sound reproduction with headphones, however, is unaffected from the listening environment acoustics. This may be considered as more exact reproduction of the intended sound. Due to smaller air volume velocities, headphones tend to also have less nonlinear distortion compared to speakers. However, the sound field produced by headphones lacks directionality as the sound field remains stationary relative to the listeners head. [2, 451]

In the transfer chain, loudspeakers and headphones have the same role of transforming the electrical signal to soundwaves. A loudspeaker is typically designed to produce flat response at the distance of one meter measured in a free-field [4,493]. However, when the listener is placed in front of the loudspeaker, the response in the listeners ears is no longer flat. This is due to the filtering occurring when the soundwaves are reflecting and diffracting of the listener’s torso head and the outer ear, on their way to the ear-drum. The behaviour of this filtering is described by the Head related transfer function (HRTF). [12,8.] A method for estimating these ear signals is presented by Marko Hiipakka in his doctoral study [12]. Hiipakka’s work [12;13] is used as the theoretical basis for the individual calibration method used in Hefio Play headphones.

In conclusion for accurate spectral reproduction, headphone’s frequency response should not indeed be flat but the missing effects of the listener’s HRTF should be compensated somehow in the design of the headphone’s target frequency response.

3.2 Non-Linear Distortion

A pure sine tone has only one spectral component; the frequency of the sine wave. Every signal imaginable is constructed with sine waves of different amplitudes and frequencies. In practise reproducing perfectly pure sine tone or multiple sine tones with a headphone is not possible. Some non-linearity is always added to the sound when played back with headphones. This non-linearity is described with several types of non-linear distortions of varying complexity.

3.2.1 Harmonic Distortion

Harmonic distortion is the simplest type of non-linear distortion. Harmonic distortion occurs when harmonics, aka overtones of a sine wave, are added to the signal. These harmonics are additional sine waves with frequency of the fundamental frequency multiplied by a whole number ($N*f_0$). The “N” in the formula describes the order of the harmonic.

As mentioned, Farina’s method enables the deconvolution of the linear impulse response and impulse responses for the harmonic distortion. According to Audio Precision [1] The IEC 60268-7 standard requires the 2nd and 3rd order harmonic distortion to be measured and specified. Usually the harmonic distortion is presented as Total Harmonic Distortion (THD) value. This value is usually presented in percentage as the level of the harmonics compared to the fundamental signal.

3.2.2 Multitone Distortion

Multitone distortion or modulation distortion is a more complex type of distortion. It occurs when a stimulus with more than one fundamental spectral component is input to a system with some non-linearity e.g. headphones. It can be beneficial to measure multitone distortions as the stimulus signals used in these measurements are closer to actual signals like music.

4 Statistical Quality Control

The concept of quality and common statistical quality control (SQC) methods are presented in this chapter. Statistical methods crucial to this thesis are introduced also. The subject is covered from the perspective of headphone manufacturing and all the examples will be from this field.

4.1 Definition of Quality and its Characteristics

Montgomery [14,6] defines quality in two ways as follows: “Quality means fitness for use.” and “Quality is inversely proportional to variability.” The first definition can be divided to two aspects; quality of conformance and quality of design. In a way the first aspect is defined by the latter. Design of a product requires specifications. Quality of conformance implies how well these specifications are met in the process of turning the design into an actual product. Focusing too much on the quality of conformance has shifted the focus from the customer to fulfilling the specifications. If the quality of the design is not good enough it is impossible to manufacture a product “fit-for-use” for the customer. Problems in quality should not and cannot be dealt only in the manufacturing of product. [14,6] In the case of headphones, even if e.g. the frequency response is designed according to the standards, but if the shape of the headphone is making it uncomfortable to wear by customers, it is impossible for this headphone to become fit-for-use without changing the design.

The second definition ties quality to variability. When variability of the characteristics impacting the quality of the product decreases, the quality of the product increases. In other words; by reducing variability one can achieve quality improvement. The characters impacting the quality of a product are called simply quality characteristics or critical to quality (CTQ) characteristics. For a headphone these CTQs are e.g. frequency response, THD, coupling to ear pressure (over-ear-headphones). The kind of characteristics that can be measured are of physical type. Other types of CTQs are sensory and time orientation. Examples of sensory are colour or appearance of the headphone. Those characteristics are harder to measure and are more to do with the preferences of the customer. Time orientation means the reliability, durability and serviceability of the product. For example, the number of times the headphone cable can be bent before the electrical connections break deeming the headphone unusable. Or the possibility to exchange the cable to a brand new one, hence pro-longing the life time of the overall product. [14,6-8]

4.2 Statistics

This section of the thesis introduces the three important numerical measures useful for describing variability. These measures are part of statistics which is the science of data analysis and conclusions based on the data [14,64].

a) Sample Average

Sample average is the arithmetic mean of the observations in a sample. It is a measure of central tendency in the sample. [14,69] The observations could be for example the average sensitivity of a headphone within a set bandwidth. In this thesis the sample consists of eight prototype earphones.

$$\bar{x} = \frac{x_1 + x_2 + \cdots + x_n}{n} = \frac{\sum_{i=1}^{n} x_i}{n} \qquad (2)$$

$x_1, x_2, \ldots, x_n$ are the observations

n is the number of observations

The sample average can be thought to represent the balancing point or the center of mass of the sample [14,69].

b) Sample Variance

Sample variance is measure of the variability in the sample. It is defined as the deviations of the individual observations from the sample average, divided by the number of observations minus one. [14,69-70]

$$s^2 = \frac{\sum_{i=1}^{n} (x_i - \bar{x})^2}{n-1} \qquad (3)$$

x_i are the observations

$\bar{x}$ is the sample average

n is the number of observations

As the deviations are squared, also the unit of the original data gets squared. This can many times be hard to interpret. [14,70] Which brings us to the next and last numerical measure of the section.

c) Sample Standard Deviation

As the sample variance is not intuitive to use, standard deviation is more often used instead. It is simply the square root of the sample variance *s^2*.

$$s = \sqrt{\frac{\sum_{i=1}^{n}(x_i - \bar{x})^2}{n-1}} \tag{4}$$

x_i are the observations

$\bar{x}$ is the sample average

n is the number of observations

The main benefit of using the sample standard deviation over the sample variance is that it has the same unit as the original measurement data or the observations.

4.3 Statistical Process Control

Statistical process control (SPC) is a set of problem solving tools targeted to reduce variability and to achieve process stability. SPC has seven major tools which are called “The magnificent seven” by Montgomery [14,180]. Those are:

1. Histogram or stem-and-leaf plot
2. Check sheet
3. Pareto chart
4. Cause-and-effect diagram
5. Defect concentration diagram
6. Scatter diagram
7. Control chart

Out of these seven only the control chart is in the scope of the thesis. The other six tools are described by Montgomery [14,199 -205].

A process has to be stable or repeatable to fulfil customer expectations. It must also operate with little variability in the quality characteristics.

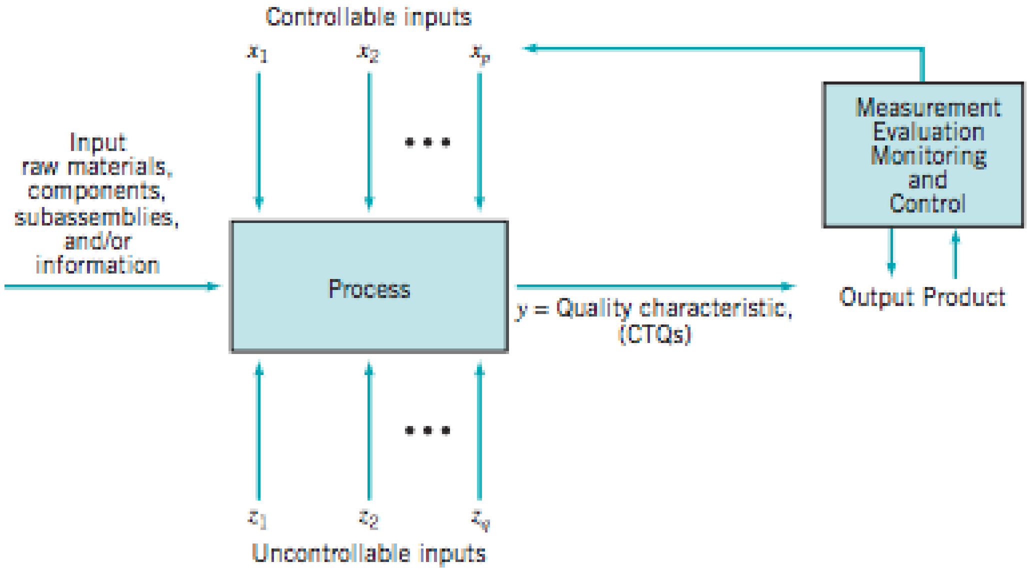

Figure 8 - Production process inputs and outputs reprinted from Montgomery [14,13]

Figure 8 shows a process and its inputs and outputs. The process in the case of headphones would be the manufacturing process. The controllable input parameters (x_1, x_2, …, x_p) are also called process variables. Pressures, temperatures and feed rates are few examples of these. The uncontrollable inputs (z_1, z_2, …, z_q) are either impossible or difficult to control. Factors such as quality of the raw materials and environmental factors fall into this category. The output (y) is a variable quality characteristic. It is a measure of both; the product and process quality A finished product has several quality characteristics. Finally, Figure 8 shows a feed-back-loop which feeds measurement, evaluation, monitoring and control information back in to the controllable inputs of the manufacturing process in order to improve quality. [14,13]

In statistical process control a process can have two states. The first state is called “in statistical control”. This means that the process has only chance causes of variation (Figure 8: uncontrollable inputs). The chance causes of variation are unavoidable and can be thought as noise. Every process will always have the chance causes of variation. The other state is called “out-of-control process”. If a process has any other kind of variability it is thought to be out of control. These causes of variation are called assignable causes of variation. Poor adjustment or control of manufacturing machines, operator errors and faulty raw materials are the three major sources of assignable cause of variation.

The main objective of SPC is to quickly detect the presence of assignable causes of variability. In other words; to quickly detect if a process gets out of control in order to minimize the number of non-conforming products produced. In conclusion eliminating avoidable variability in the process is the ultimate goal of Statistical process control. [14,181.]

4.3.1 Control Chart

Control chart is primary technique for on-line process monitoring. It was invented by Walter A. Shewhart of the Bell Telephone Laboratories in the 1920s [14,180]. Figure 9 shows an example control chart.

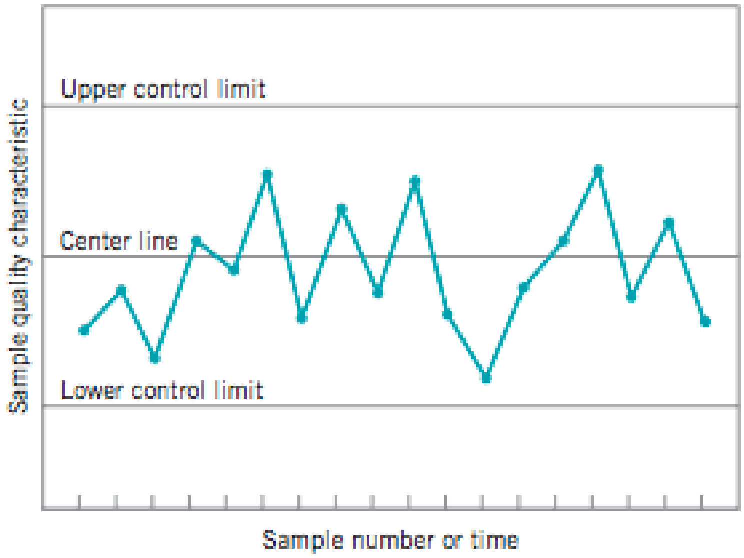

Figure 9 - Typical Control Chart. Reprinted from Montgomery [14,182].

Control chart plots the measured or computed values of a quality characteristic as a function of time or sample number. In Figure 9 the x-axis shows the sample number or time and the y-axis shows the value of the sample quality characteristic. Usually control charts are applied to the output variable of the process but sometimes it can be useful when applied to inputs instead. Control chart like in Figure 9 has three horizontal lines. Center line represents the average value of the CTQ calculated when the process in statistical control. The other two lines are the upper control limit (UCL) and the lower control limit (LCL).

Unusual variability can move the plot outside the limits. Given that these limits are set correctly, this indicates that the process is out of statistical control. Another indicator of an out-of-control process is when the pattern of the chart acts non-randomly. The scenario is possible even if the plot stays within the set control limits. For example, if many consecutive points plot between the LCL and the center line it would suggest something is wrong in the process. In conclusion the plot of the chart should form a random pattern mostly within the control lines indicating the process is in statistical control. [14,182-183]

However, control chart is vulnerable to two types of errors. Error type 1 is the scenario when the process is interpreted to be out of control when it really is in control. Error type 2 is the opposite of type 1; process is thought to be in control when it really is out of control. Setting the control limits is a trade of between type 1 and type 2 errors. When the control limits are moved farther from the center line risk of a type 2 error occurring increases. Simultaneously the risk of type 1 error decreases. Logically the opposite effect is achieved when the control limits are moved closer to the center line. [14,183 189.] A general model for setting the control limits is presented by Montgomery [14,185]. The model is presented in the Equation 5 below as it is in the source material [14,185]. The w in the equation stands for a sample statistic that measures a CTQ.

$$
\begin{aligned}
UCL &= \mu_w + L\sigma_w \\
Center\ line &= \mu_w \\
LCL &= \mu_w - L\sigma_w
\end{aligned}
\tag{5}
$$

μ_w is the mean of w.

L is the distance of the control limits from the center line.

σ_w is the standard deviation of w.

It is important to note that control chart is used only to monitor a process. Its main purpose is to indicate assignable causes of variation in a process. The needed actions to eliminate the causes of variability are not suggested by the control chart alone. [14,185] A control chart will be used in chapter 7.2 for plotting the average sensitivity of the sample headphones within a selected frequency band.

4.4 Designed Experiment

Alongside statistical process control, designed experiment is very powerful tool used to improve a process. Like the last section points out, SPC is only used for monitoring purposes. Thus, SPC is a passive statistical quality control method, whereas designed experiment is an active one.

Designed experiment helps to find the most deterministic variables affecting the quality characteristics. The controllable input factors, shown in Figure 8, are systematically varied to determine the effect of these factors to the output of the process. The designed experiment reveals which controllable inputs have the biggest impact on the quality and where to set those inputs so that the variability of the output is small, the effects of the uncontrollable inputs get diminished and the output is near its required target. Designed experiment is invaluable in lowering the variability in CTQ characteristics. It aims to actively improve the manufacturing process.

Whereas SPC is a passive on line quality control tool, designed experiment is an active off line tool. It is especially effective when used in the early stages of manufacturing and development of a product. It can also help to establish the statistical process control for the manufacturing process. For example, it can help to set the control limits of a control chart. Benefits of applying designed experiments early in a process can lead to reduced development time and cost, reduced variability and thus improved quality, as well as improved yield. Yield is the number of working units produced divided by the total number of produced units. [14,550-551.]

4.5 Acceptance Sampling

Acceptance sampling is one of the oldest methods of quality assurance. It is the inspection of sample from a lot. The lot could contain a sub-assembly from a contractor, e.g. headphone drivers or even the final product, as in this thesis, the headphones. A preferably random sample is taken from the lot and inspected for certain quality characteristics. Decision is made to accept or reject the lot based on the inspection of the sample. This procedure is also called lot sentencing. The rejected lots can be then either scrapped or returned to the sub-contractor or re-worked some other way.
Usually acceptance sampling is not used to estimate the quality of the lot but to simply sentence the lot based on the sample. In the worst-case-scenario this can lead to rejecting a “good” lot or accepting a “bad” lot. Other options for lot sentencing are to accept the lot without inspection and inspection of the whole lot. Decision between these three alternatives is mainly a question of time and cost. Acceptance sampling is less emphasized in the modern quality assurance systems. Statistical process control and designed experiments are more commonly used and provide actual feedback on the manufacturing process that can lead to quality improvement. [14,15-16,632-636.]

Since the Hefio Play headphones are still in the prototyping phase, all the assembled prototypes are inspected. Acceptance sampling will become a useful method only during the mass production of the headphones.

4.6 Quality Control in Headphone Manufacturing

The definition of an earphone quoted from Audio Precision [1] in chapter 2.2 shows that an earphone is basically a small loudspeaker coupled close to the ear. Thus, describing quality control of loudspeaker manufacturing translates quite well to headphones.

Chapter 3 of this introduces some measurable metrics of headphones. These metrics are also quality characteristics. Frequency response, distortions and maximum SPL are examples of many possible metrics used to describe loudspeaker performance. The magnitude of the frequency response is probably the most measured metric of loudspeakers. Typically, a manufacturer will use a “golden” or “reference” sample to quantify these metrics. It is then communicated to customers with presumption that all of the produced units will have similar characteristics as the reference sample However, tolerance in the process and parts shifts every unique unit away from both the reference sample and any other unit. [15,1-2.]

Goldberg suggests in his paper [15,2] a new metric that is missing from current standards for loudspeaker measurements. This metric is the spread of the frequency response data. Basically, each measured unit is compared to the average response of the whole sample. A side note on the tightness of the spread: an active design can tighten the spread of a loudspeaker. This suggests that the DSP-unit of the Hefio Play headphones could improve the spread.

5 Measurement System

5.1 Functionality

The measurement unit in its core is an audio playback and record system. For the purposes of the study the response data is recorded with ARTA – Audio Measurement and Analysis software [16]. The actual measurement device to be used in the production will be implemented with a MATLAB-script running on a standard PC. The basic operation goes so that first the measurement signal is output to the headphone through the audio interface. That signal, played through the headphone, is then recorded with the measurement mic back through the audio interface to the computer. This process is visualized in Figure 10.

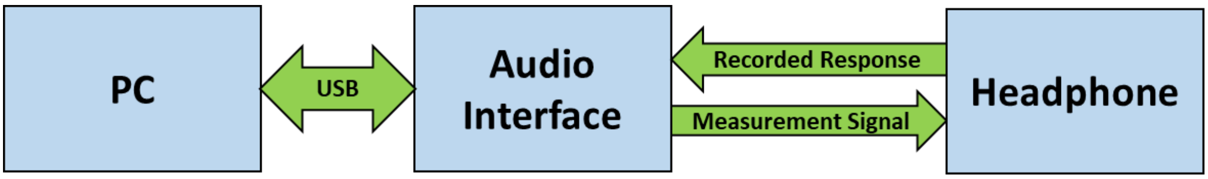

Figure 10 - Flowchart of the measurement process.

The process described above is quite simple in its core. The actual implementation will have more complexity due to the acoustic properties of the unit and the automation of the data collection. The following sections concentrate on the physical implementation of the measurement device.

5.2 Equipment

5.2.1 Microphone

The microphone used for all the measured conducted for the study is the Knowles FG-23329-P07 [17]. It is an omnidirectional electret condenser microphone with frequency range from 100 Hz to 10 kHz. An additional bias circuit was used to supply the microphone with the needed bias voltage.

5.2.2 Audio Interface

The audio interface used for the measurements is the Tascam US-16x08 [18]. One balanced input channel (input 1) and the phones output is utilized in all the measurements. Sample rate is set to 48 kHz and the resolution to 24-bit.

5.2.3 Tube & Adapter

Copper measurement tube and the adapter for connecting the earphone into the tube are pictured in Figure 11. The length of the copper tube is ten meters. The length is more than enough to attenuate the reflections of the end of the tube to be negligible in the impulse response measurements.

Figure 11 – Picture of the copper measurement tube and the 3D-printed adapter.

The earphone side of the adapter is designed to tightly fit the tip of the earphone to minimize the leakage between the surfaces. This assures that most of the signal is captured inside the tube. The cross-section area of the copper tube was picked to be as close as possible to the cross-section area of the opening of the earphones sound hole. This minimizes the reflections from the transition area from the adapter to the tube. However, the copper tube was bought off the shelf of a hardware store, so an exact match was not available.

The cross-section area transitions gradually from the slightly larger cross-section area of the copper tube to the smaller cross-section area of the earphone’s sound hole Inside the thinner piece of the adapter. Measurements conducted with the adapter show no signs of interfering reflections.

5.2.4 Computer

The computer used for the measurements (ARTA [16]) and the analysis (MATLAB) was the HP 250 G4 Notebook PC. It has Intel® Core™ i3-5005U CPU clocked at 2 GHz with 4 GB of RAM. The operating system running on the PC was the 64-bit version of Windows 10 Home.

6 Measurements

The measurement setup, the measured sample earphones, and the measurement data and its analysis are presented in this chapter. Two sets of measurements were conducted for this thesis. The sets of measurements are described shortly below and in more detail in section 6.4. of the chapter.

1. Sample Frequency Response

The impulse responses of eight prototype earphones were measured. The equipment and measurement setup described in section 5.2. of Chapter 5 and in section 6.1. of this chapter were used.

2. Repeatability Measurement

The impulse response of PROTO4R earphone was measured ten repetitive times. The equipment and measurement setup described in section 5.2. of Chapter 5 and in section 6.1. of this chapter were used.

6.1 Measured Earphones

It must be noted that the eight prototype earphones are only the passive earphones measured completely without the DSP-unit described in chapter 2.6. of this thesis. The measurement signal was played back through only the woofer driver. The chassis of each earphone was 3D-printed, and the earphones were hand assembled.

The original purpose of the prototype earphones used in this study was to develop and test various acoustical features, which means that the prototypes can be expected to have dissimilar frequency responses. Mass produced headphones, however, would be required to have the smallest possible frequency response spread between samples.

6.2 Measurement Signal

The same logarithmic sine sweep measurement signal was used in all the measurements. The signal was generated in the ARTA software. According to the user manual of ARTA [16,74] the swept sine used in ARTA is based on Farina's definition presented in Chapter 3.1. of the thesis in Equation 1.

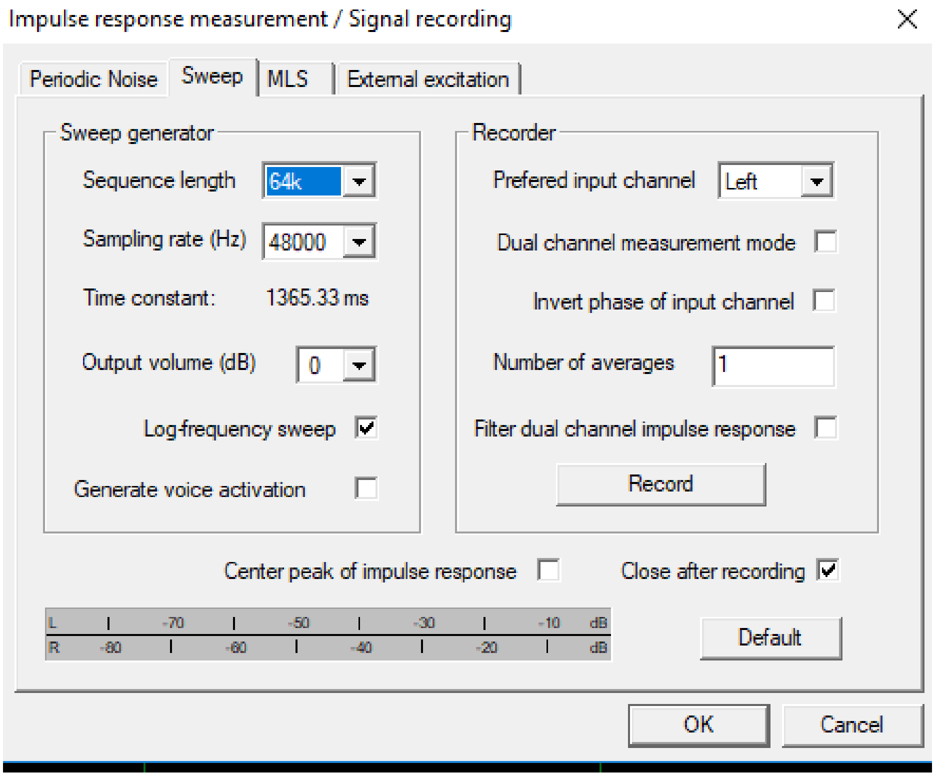

Figure 12 - Impulse response measurement parameters. Screen capture from the setup of ARTA software.

Figure 12 shows the setup window of the impulse response measurement in ARTA. The length of the measurement signal was set to 1365,33 milliseconds with a sample rate 48 kHz. Number of averages was set to ten. The number of averages sets how many times the sweep is played back. The measured impulse response is then the average response of those repetitions. [16.]

6.3 Measurement Data

The measurement data from both sets of measurements is plotted in frequency domain and discussed in this section. Also, the pre-processing of the data is explained.

6.3.1 Sample Frequency Responses

The impulse responses of the eight prototype earphones were measured using ARTA software. Each prototype's impulse response was averaged from ten consecutive logarithmic sweeps. The impulse response data was then exported as ASCII-files. The frequency responses were then calculated, from the impulse response data, and plotted with MATLAB-script.

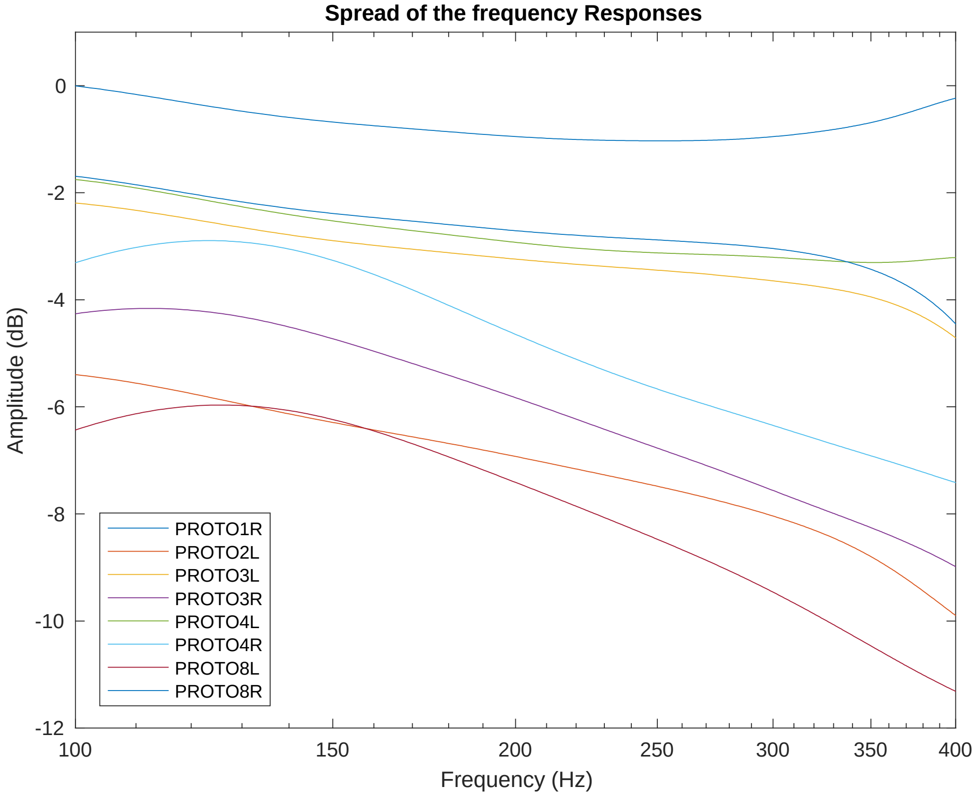

Figure 13 - Spread of the frequency responses normalized to the highest amplitude.

Figure 13 above shows the plotted frequency responses of the eight prototype earphones. For the purposes of this thesis the bandwidth of all the measurements was limited from 100 Hz to 400 Hz. Since the measured eight prototype earphones are early handmade prototypes the shapes of the full band width (20 Hz – 20000 Hz) curves were hard to compare and the deviation in the shape would distort the deviation in the amplitude. Instead a somewhat uniformly shaped portion of the spectrum was used in the analysis. The calculated sample frequency response data was normalized to the maximum amplitude in the data within the considered band.

6.3.2 Repeatability Measurement

The impulse response of PROTO4R earphone was measured ten times using ARTA software. Between every individual repetition the earphone was reattached to the adapter described in section 5.2.3. All the repetitions were measured one after another in similar environment by a single operator. The impulse response data was then exported in ASCII file format from ARTA. The frequency responses were calculated and plotted with MATLAB-script from the impulse response data.

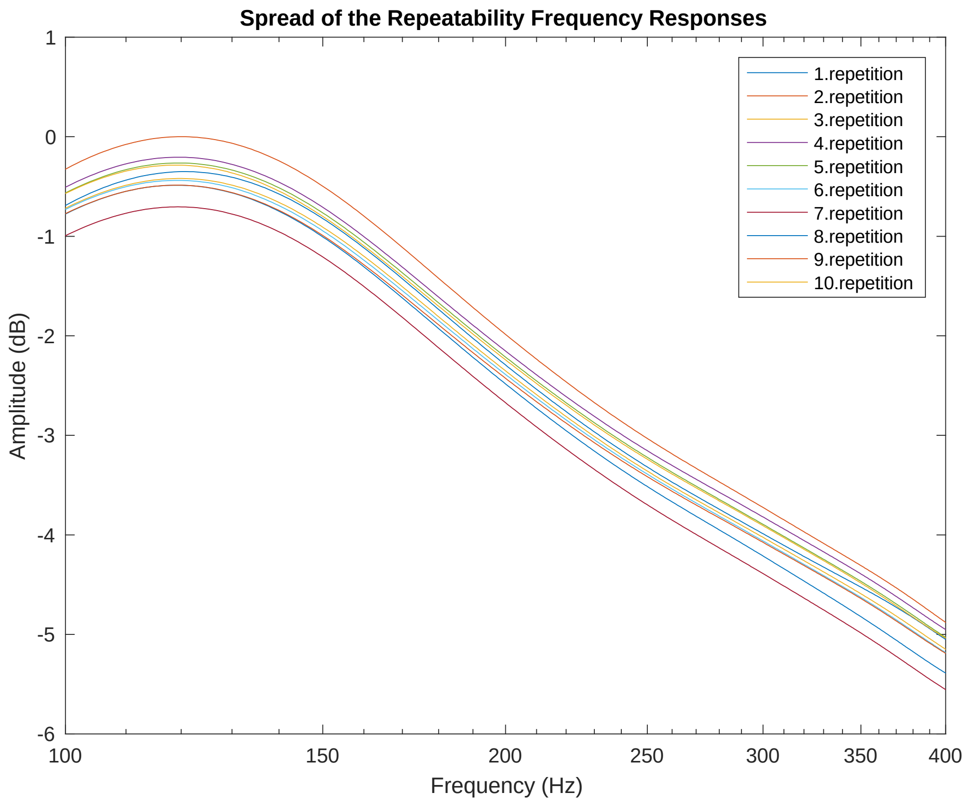

Figure 14 - Spread of the repeatability measurement normalized to the highest amplitude in the data.

The ten repetitions of the frequency response of the PROTO4R earphone are plotted in Figure 14. The bandwidth of 100 Hz to 400 Hz was used again here to have the measurements comparable to the sample frequency response measurements. The repeatability frequency response data was normalized to the maximum amplitude in the repeatability measurement data within the considered band.

7 Analysis

This chapter is focused in the analysis of the measurement data presented in the section 6.5 of Chapter 6. The statistical quality control and numerical methods presented in sections 4.2. and 4.3. of Chapter 4 are utilized in the analysis. Metric and methods for quantifying loudspeaker consistency, described by Goldberg [15], are also utilized.

7.1 Sample Average Response

Sample average response was calculated from the sample frequency responses. The sample average response curve was obtained from the sample frequency response data by calculating the mean of the amplitude in each frequency point.

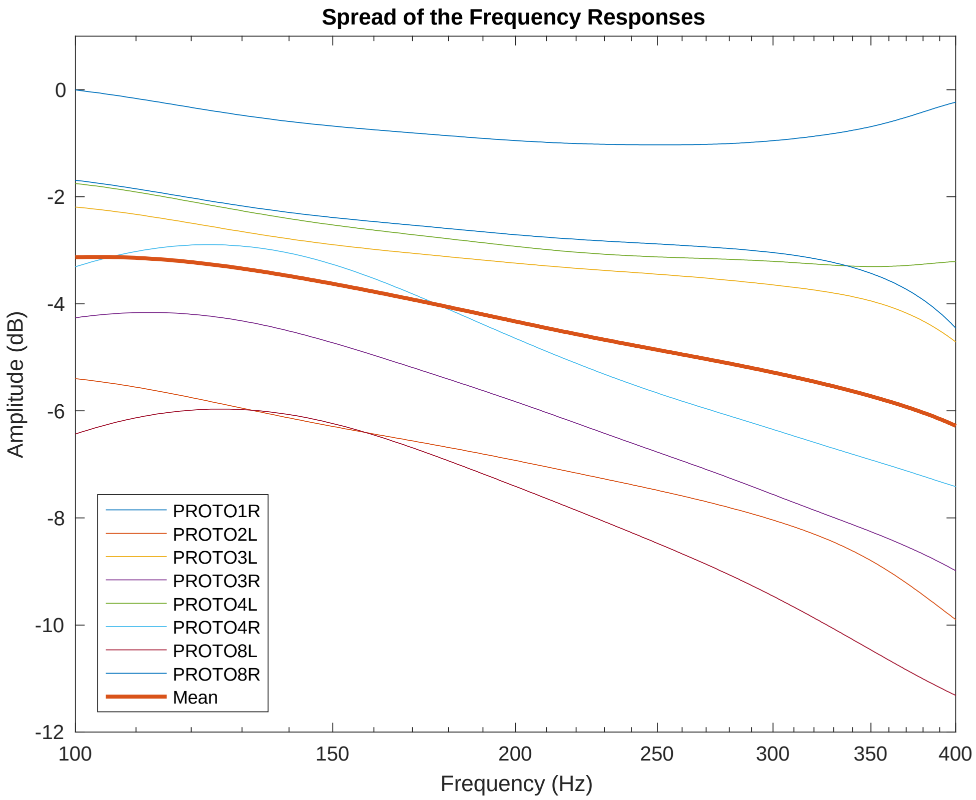

Figure 15 - Sample frequency responses and the average frequency response (Mean) plotted with MATLAB.

Figure 15 shows the plotted sample frequency response data plus the thicker sample average response curve (Mean in Figure 15). From this curve one can easily see the shape of the sample average response.

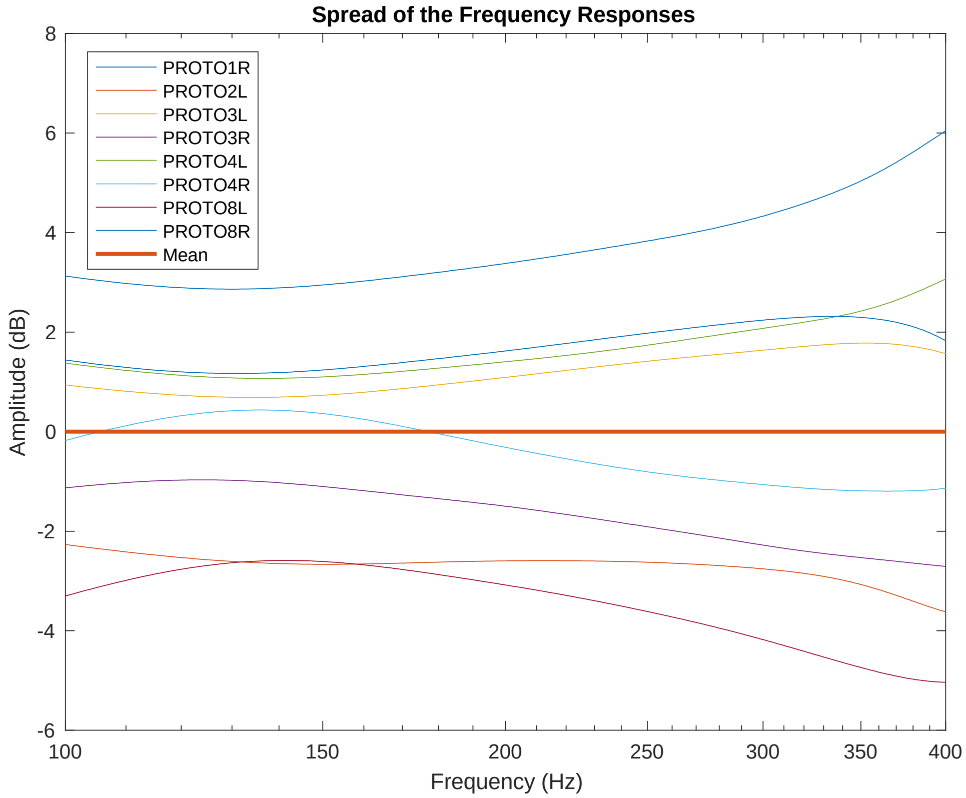

Figure 16 - Spread of the frequency responses normalized to the average response curve (Mean). Plotted with MATLAB.

Figure 16 shows the same data as Figure 15 but this time the sample frequency response data is normalized to the sample average response curve. This kind of curve visualizes the spread of the individual sample responses from the sample average response curve better. The idea for the representation style was obtained from Goldberg [15,4].

In Goldberg's paper [15,8-9] loudspeaker 3 is rated average (E) The spread of its response data is 3.4 dB from maximum to minimum value. The poorest rating in his paper is terrible (H) with the spread of the response data being more than 4.5 dB. (Table 7 in Goldberg's paper) In Figure 16 the spread of the response data from maximum to minimum is 11.3 dB.

However, Goldberg also states that the sample size in the analysis should be at least 100 units which are produced within minimum of one year [15,4]. Thus, a larger sample size is needed to get accurate results on the quality of the produced Hefio Play earphones, if this kind analysis will be used.

7.2 Average Sensitivity

The average sensitivity of each prototype was calculated with a MATLAB script. In practise the arithmetic mean amplitude of each prototype earphone was calculated within the specified frequency band (100 Hz - 400 Hz). This particular arithmetic mean value is called the average sensitivity in the thesis.

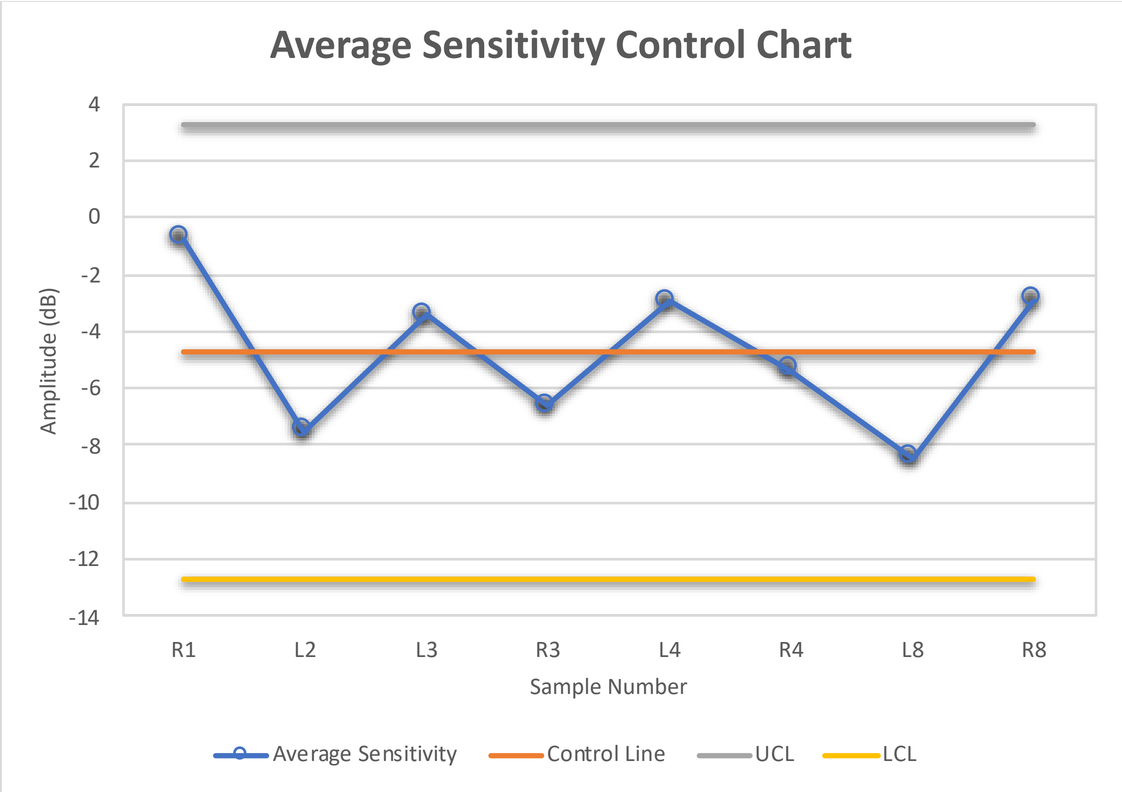

Figure 17 - Average sensitivity from 100 to 400 Hz of each prototype earphone plotted in a control chart.

The dots along the blue curve in Figure 17 plot the average sensitivity, of each prototype earphone (R1, L2, L3, R3, L4, R4, L8, R8) within the considered band from 100 Hz to 400 Hz. The y-axis of the control chart is the amplitude in dB normalized to the same maximum amplitude as in Figure 13. In addition to the plot of the average sensitivities the control chart has the center line and the upper and lower control limits.

The distance of the control limits from the center line was calculated using Equation 5 presented in section 4.3.1. of Chapter 4. The *L* in Equation 5, being the distance from the sample average in standard deviation units, was chosen to equal three. According to Montgomery [14] in the United States, it is customary to set the control limits at the distance of three times the sample standard deviation from the center line. Limits set at that particular distance from the center line are called three-sigma limits [14,190].

The principles of the control chart in Figure 17 could be used as the basis for the future statistical process control of the mass produced Hefio PLAY headphones. However, a larger sample size is needed in order to set the center line and the control limits more accurately as the section 7.1 already implies.

Table 1 - Statistical numerical measures of the average sensitivity sample.

Numerical Measure	Value (dB**)**
Sample Average	-4,745
Sample Variance	7,091
Sample Standard Deviation	2,663

Sample average, sample variance and sample standard deviation of the average sensitivity data were calculated with a MATLAB script. The results of those statistical numerical measures are shown in Table 1 above.

7.3 Repeatability

The purpose of the repeatability measurement is to find out the accuracy of the measurement conducted with the setup presented in section 6.3. of this chapter. The better the measurement accuracy is, the tighter a cluster the consecutive measurements of the same earphone should form.

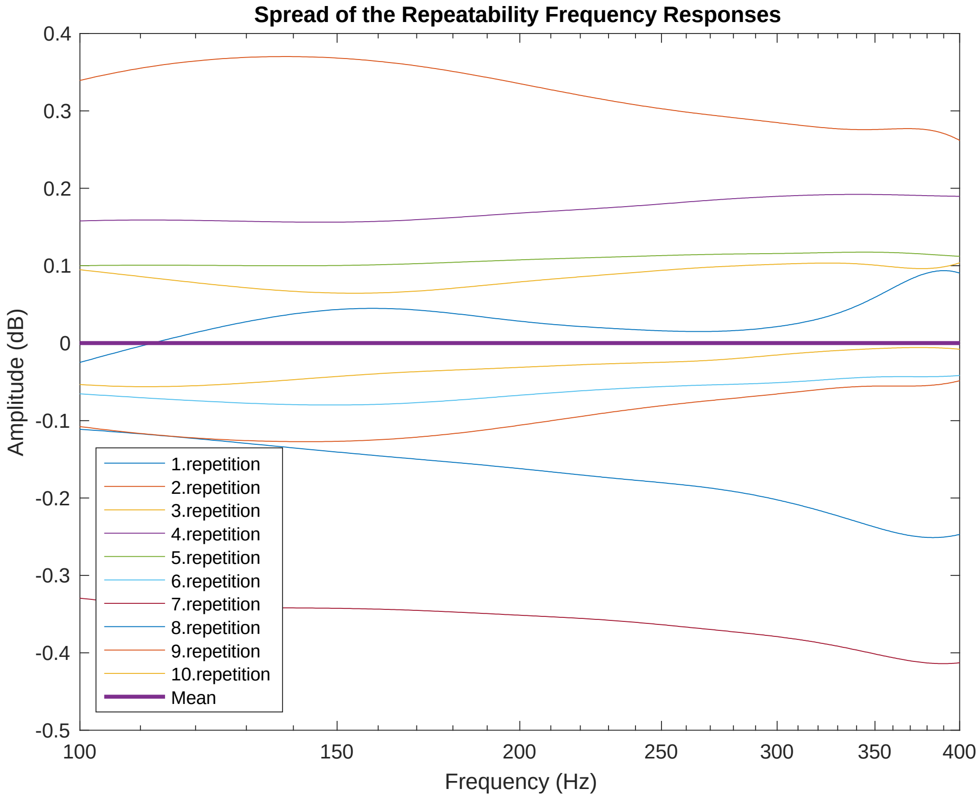

Figure 18 - Spread of the repeatability responses normalized to the average response (Mean) of the repetitions.

For better visualization of the spread the plot in Figure 18 was created in MATLAB. The plot shows the responses of the repetitions normalized to the average response curve. The average response curve was obtained by calculating the mean of the amplitude in each frequency point across the ten repetitions. The idea for this type of a representation was obtained from Goldberg [15,4].

Table 2 -Statistical numerical measures of the repeatability measurement sample.

Numerical Measure	Value (dB)
Sample Average	-2,976
Sample Variance	0,037
Sample Standard Deviation	0,192

The average sensitivity across the bandwidth of the repeatability data was calculated for each repetition. Sample average, sample variance and sample standard deviation of that average sensitivity data were calculated with a MATLAB script. The results of those statistical numerical measures are shown in Table 2 above.

7.4 Measurement Accuracy

Measurement accuracy sets the smallest possible measurable difference of the measured metric. Table 3 shows sample variance, sample standard deviation and sample maximum and minimum values of both the average sensitivity and the repeatability data.

Table 3 - Comparison of the statistical numerical measures of the average sensitivity and the repeatability data.

Numerical Measure	**Average Sensitivity**	**Repeatability**
Sample Variance	$\sigma_a^2 = 7.091$ (dB$^{2)}$	$\sigma_r^2 = 0.037$ (dB$^{2)}$
Sample Standard Deviation	$\sigma_a = 2.663$ (dB)	$\sigma_r = 0.192$ (dB)
Sample Max	$a_{max} = -0.750$ (dB)	$r_{max} = -2.662$ (dB)
Sample Min	$a_{min} = -8.455$ (dB)	$r_{min} = -3.345$ (dB)

The results in Table 3 indicate that the spread of the average sensitivity data is much wider than the spread of the repeatability data. From the data in Table 3 one can calculate the precision-to-tolerance (P/T) ratio. This ratio is used to compare the precision of the measurement (repeatability), aka gauge capability, to the tolerance band of the measurement (average sensitivity) [14,370 - 371]. Equation 6 shows the formula for calculating the P/T ratio [14,370].

$$P/T = \frac{6\sigma_r}{a_{max} - a_{min}} = 0.150 \qquad (6)$$

σ_r is the sample standard deviation of the repeatability data.

a_{max} is the maximum value of the average sensitivity data

a_{min} is the minimum value of the average sensitivity data.

According to Montgomery [14,371] P/T ratio of equal or less to 0.1 implies that the precision of the measurement is acceptable. This would imply that the calculated P/T value of 0.15 is close to acceptable. However, the wide spread in the average sensitivity data is distorting the P/T value to “look better”. As the spread of the produced earphones gets tighter the precision of the measurement stays the same. This results in larger P/T value as the consistency of the produced units gets better. Thus, the calculated P/T value is not an accurate representation of the measurement accuracy. A bigger and more consistent sample is needed also here to assess the capability of the measurement device more accurately.

8 Conclusions

The goal of this study was to present the proof of concept of an acoustic measurement system. The study succeeded to proof the concept of measuring the Hefio Play headphones with the method described in this thesis. However, due to the small amount of measured earphone prototypes and the wide spread of their frequency responses, the study fails to fully evaluate the performance of the measurement system. Further testing with larger sample size is needed.

The measurement and analysis methods described and tested in the study should work on larger sample size without any modifications. However, further improvement of the measurement accuracy will be needed when the variability of the earphones gets smaller to maintain an acceptable P/T ratio.

The measurements conducted for this study reveal only that there is a lot of variability in the prototype earphones. The source of the variability remains unknown. Introducing designed experiments to the process could be the most effective way to find the causes of the variability. It would be most beneficial to repeat the measurements described in this thesis only after the consistency of the earphones is within reasonable limits.

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